AudioCodes™

MediaPack MP-118 & MP-114

Quick Installation Guide

Integration with Microsoft® Office Communications Server 2007

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Reader's Notes

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Notices

Notice

This Quick Installation Guide describes the installation and configuration of AudioCodes' Media Pack MP-11x FXO SIP analog media gateways for integration with Microsoft® Office Communications Server 2007.

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Each abbreviation, unless widely used, is spelled out in full when first used, and only Industry standard terms are used throughout this manual.

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Related Documentation

Document #	Manual Name
LTRT-656xx (e.g., LTRT-65601	MP-11x & MP-124 SIP Release Notes
LTRT-654xx	MP-11x & MP-124 SIP User's Manual

Warning: Ensure that you connect the FXO ports to Central Office (CO) / PBX lines only.

Warning: The MediaPack is supplied as a sealed unit and must only be installed or serviced by qualified service personnel.

Warning: Disconnect the MediaPack from the electrical mains and from the Telephone Network Voltage (TNV) before servicing.

Note: MediaPack and MP-11x refers collectively to the MP-118 and MP-114 FXO VoIP gateways.

Note: Where "network" appears in this manual, it means LAN, WAN, etc. accessed via the gateway's Ethernet interface.

Note: **FXO** (Foreign Exchange Office) is the interface replacing the analog telephone and connects to a Public Switched Telephone Network (PSTN) line from the Central Office (CO) or to a Private Branch Exchange (PBX). The FXO is designed to **receive** line voltage and ringing current, supplied from the CO or the PBX (just like an analog telephone). An FXO VoIP gateway interfaces between the CO/PBX line and the Internet.

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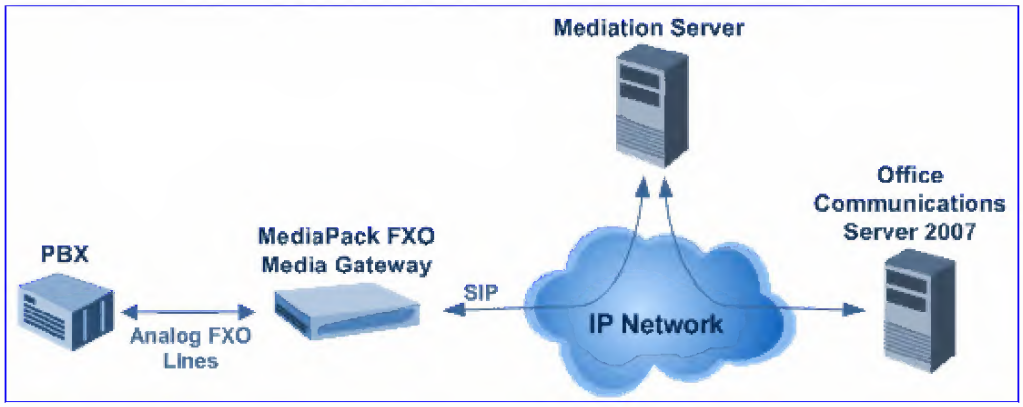
1 Introduction

This guide provides step-by-step instructions for quickly setting up AudioCodes' MediaPack (MP-118 and MP-114) Session Initiation Protocol (SIP) voice-over-IP (VoIP) gateways for intermediating between third-party, private branch exchanges (PBX) or various Public Switched Telephone Network (PSTN) interfaces, and Microsoft® Office Communications Server 2007 (*Office Communications Server 2007*). For further information regarding Office Communications Server 2007, refer to Microsoft's Web site (http://www.microsoft.com/uc).

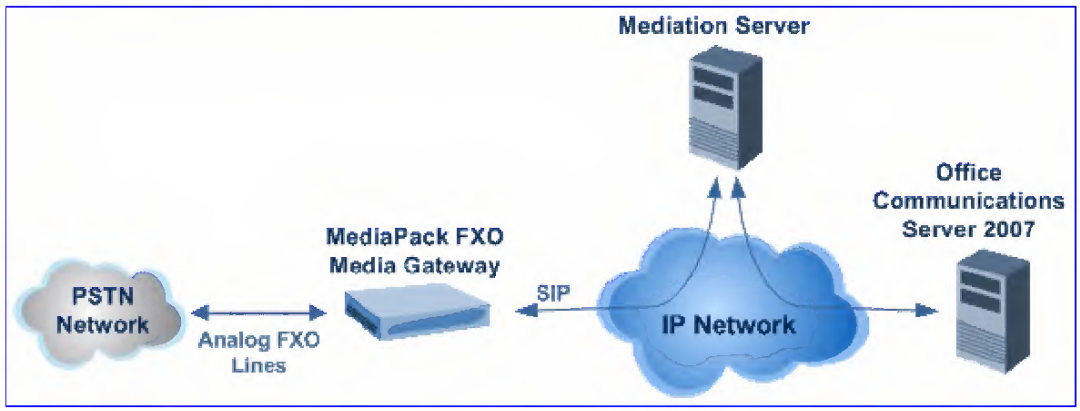
The figure below illustrates a network architecture with a PBX. In this scenario, the AudioCodes' gateway connects to the PBX using analog lines (FXO), and to Office Communications Server 2007 through Microsoft® Mediation Server (*Mediation Server*) on the IP network. The MediaPack gateways ensure interoperability with leading, third-party PBXs.

Figure 1-1: Network Architecture with PBX



The figure below illustrates a network architecture that provides a direct connection to the PSTN network. In this scenario, the AudioCodes' gateway connects to PSTN using analog lines. At the other end, the gateway connects to Office Communications Server 2007 through Mediation Server on the IP network.

Figure 1-2: Network Architecture with PSTN Interface



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The AudioCodes' MP-11x FXO gateway can be integrated in the Office Communications Server 2007 / Unified Communications (UC) environments, by implementing one of the following options:

- **One-to-One:** each PBX line connected to an FXO port on the gateway, serves one Office Communications Server 2007 user / endpoint (*OCS client*) and is defined for automatic dialing for this OCS client. A PBX or PSTN user that wants to call the OCS client, dials the OCS client's PBX line number.
- **One-to-Many:** group of PBX lines connected to the gateway serves all the OCS clients. When a new call is received from the PBX, the gateway answers the call and performs one of the following:
  - Plays a dial tone, after which the caller can dial the OCS client's extension number.
  - Plays a pre-recorded greeting message to the caller, requesting the caller to dial the desired OCS client (for example, "Welcome company X, please dial the desired extension number"). The gateway acts as an interactive voice response (IVR) system. For this option, you may need to upload a pre-recorded tone file (refer the *User's Manual*).
- **PBX-Sends-Digits:** group of Direct Inward Dialing (DID) PBX lines connected to the gateway serves all the OCS clients. When a new call is received from the PBX, the gateway answers the call, and then the PBX automatically sends the OCS client number as it was received.

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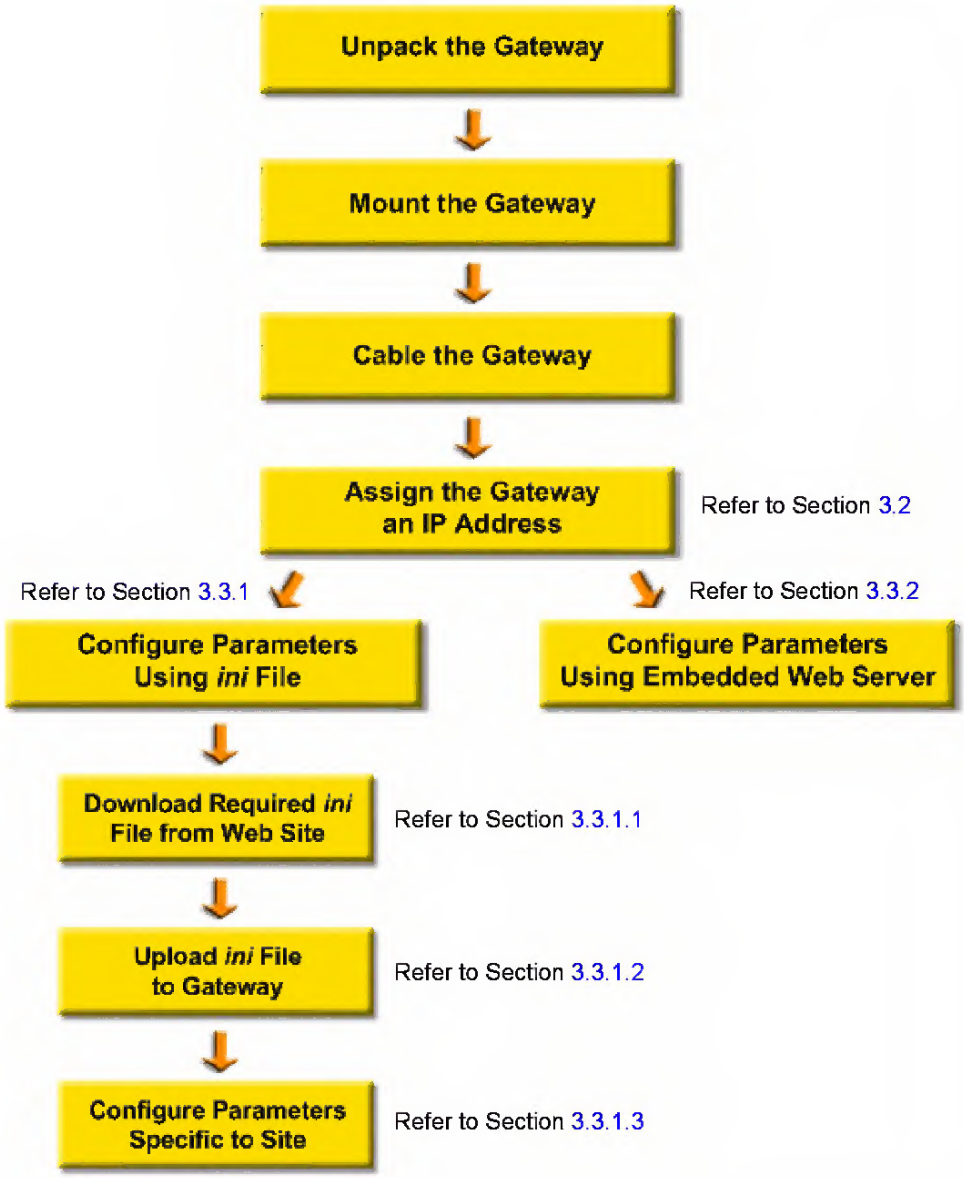
2 Quick Start

This Quick Installation Guide helps you setup the MediaPack gateways for interoperability with either third-party PBXs or various PSTN interfaces, and Office Communications Server 2007. Prior knowledge of IP networks is recommended.

Figure 2-1: Required Steps to Install MediaPack


Unpack the Gateway
Mount the Gateway
Cable the Gateway
Assign the Gateway an IP Address
Refer to Section 3.2
Refer to Section 3.3.1
Refer to Section 3.3.2
Configure Parameters Using ini File
Configure Parameters Using Embedded Web Server
Download Required ini File from Web Site
Refer to Section 3.3.1.1
Upload ini File to Gateway
Refer to Section 3.3.1.2
Configure Parameters Specific to Site
Refer to Section 3.3.1.3

Note: For detailed information on how to fully configure the gateway, refer to the MediaPack User's Manuals.

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3 Configuring the MediaPack

The gateway is supplied with the application software residing on its flash memory (set to factory default parameters).

Note: This guide assumes that the gateway is running firmware version 5.0. If you are running an earlier version, please update the version to 5.0 (available on AudioCodes' Web site -- refer to Section 3.3.1.1 on page 17).

3.1 Embedded Web Server Management Tool

The MediaPack gateway contains an embedded HTTP-based server that provides a user-friendly, client Web interface for gateway configuration and management.

3.1.1 Accessing the Embedded Web Server

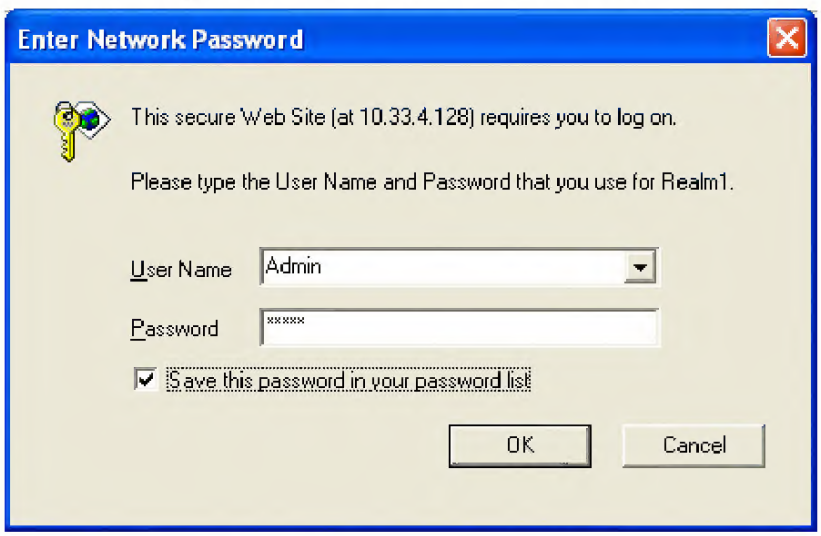
The MediaPack's Embedded Web Server is initially accessed using the default IP address (10.1.10.11), login user name ('Admin'), and login password ('Admin').

Note: Ensure that your MediaPack is in the same subnet as the PC running the Web browser. If not, refer to Section 3.2 on page 14 for assigning an IP address to the MediaPack.

➢ **To access the Embedded Web Server, take these 4 steps:**

1. Open a standard Web-browsing application such as Microsoft™ Internet Explorer™ (Version 6.0 or higher) or Netscape™ Navigator™ (Version 7.2 or higher).
2. In the Web browser's URL address field, enter the IP address of the gateway; the Embedded Web Server's 'Enter Network Password' screen appears, as shown in the figure below:

Figure 3-1: Enter Network Password Screen


Enter Network Password
This secure Web Site (at 10.33.4.128) requires you to log on.
Please type the User Name and Password that you use for Realm1.
User Name
Admin
Password
*****
Save this password in your password list
OK
Cancel

3. Enter the case-sensitive, default username and password. You can later change the login user name and password, as described in Section 3.1.3 on page 12.
4. Click **OK**; the Embedded Web Server is accessed, displaying the 'Quick Setup' screen.

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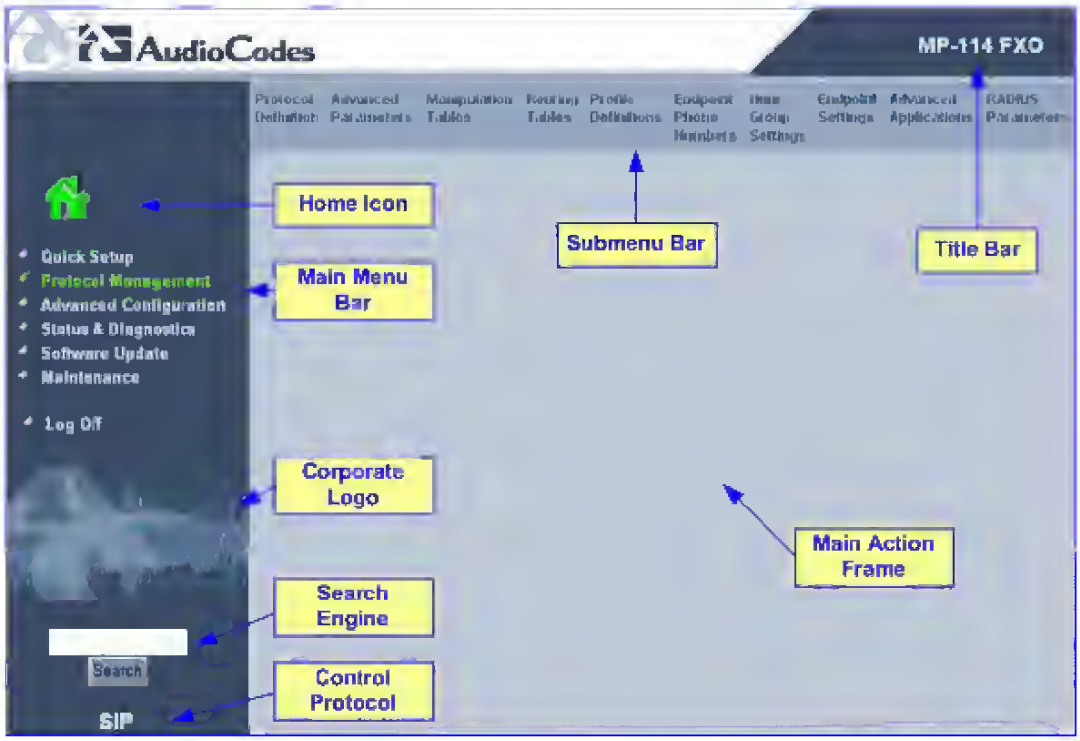
3.1.2 Navigating the Embedded Web Server

The main areas of the Web interface are shown in the figure below.

Figure 3-2: MediaPack Web Interface Areas



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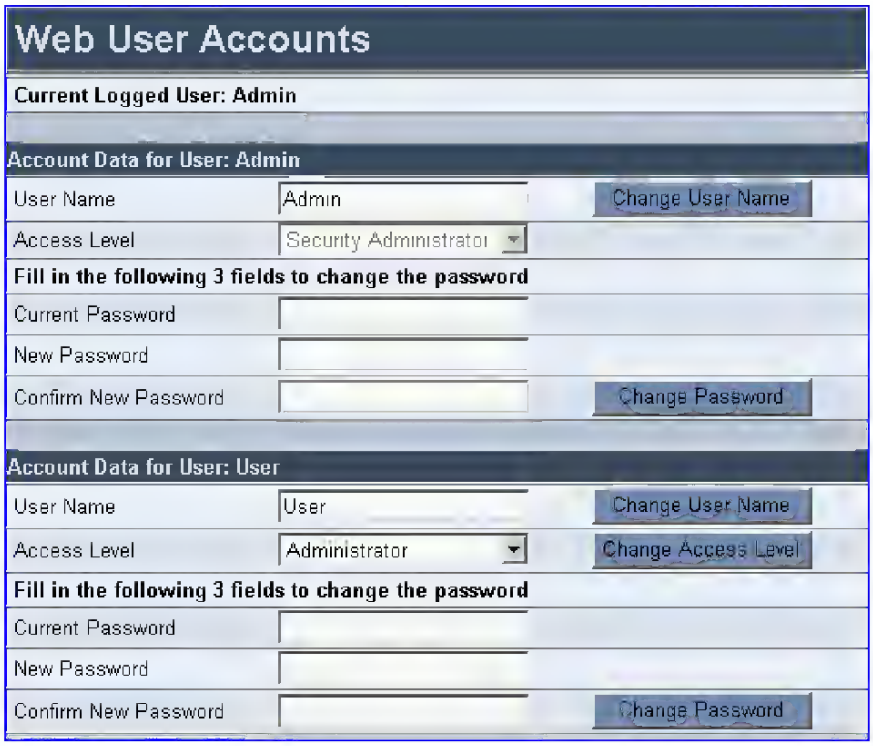
3.1.3 Changing Login Username and Password

It's recommended that you change the default user name and password of the Web user account initially used to access the Embedded Web Server.

➢ **To change the user name and password, take these 3 steps:**

1. Open the 'Web User Accounts' screen (**Advanced Configuration** menu > **Security Settings** > **Web User Accounts**)

Figure 3-3: Web User Accounts Screen



2. Change the user name by performing the following:
   a. In the 'User Name' field, enter the new user name (maximum of 19 characters).
   b. Click the **Change User Name** button; the new user name is immediately applied and the 'Enter Network Password' screen appears (shown in Figure 3-1 on page 10).
   c. In the 'Enter Network Password' screen, enter the new user name (case-sensitive).
3. Change the password by performing the following:
   a. In the 'Current Password' field, enter the current password.
   b. In the 'New Password' and 'Confirm New Password' fields, enter the new password (maximum of 19 characters).
   c. Click the **Change Password** button; the new password is immediately applied and the 'Enter Network Password' screen appears (shown in Figure 3-1 on page 10).
   d. In the 'Enter Network Password' screen, enter the new password (case-sensitive).

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3.1.4 Searching for *ini* file Parameters

The Embedded Web Server provides a search engine that allows you to search any *ini* file parameter that is configurable using the Embedded Web Server. The search result provides you with a link to the relevant Web page in which the parameter appears. You can search for a specific parameter (e.g., 'EnableIPSec') or a sub-string of that parameter (e.g., 'sec'). If you search for a sub-string, the Embedded Web Server lists all found parameters that contain the searched sub-string in their parameter names. The **Search** button, located near the bottom of the Main menu bar is used to perform parameter searches.

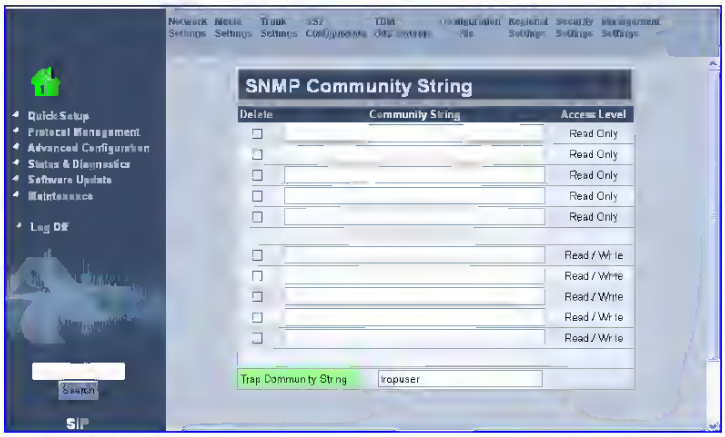
➢ **To search for an *ini* file parameter configurable by the Embedded Web Server, take these 3 steps:**

1. In the 'Search' field, enter the required name or sub-string of the *ini* file parameter.
2. Click **Search**, the 'Searched Result' screen appears listing all searched parameter results.

Figure 3-4: Searched Result Screen

3. In the Searched Result list, click the required parameter to open the screen in which the parameter appears. In the relevant screen, the searched parameter is highlighted in green for easy identification, as shown in the figure below.

Figure 3-5: Searched Parameter Highlighted in Screen



Note: If the searched parameter is not located, the "No Matches Found For This String" message is displayed.

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3.2 Assigning an IP Address to the MediaPack

If the gateway's default IP address (refer to Table 3-1) doesn't correspond to the network in which Office Communications Server 2007 is installed, you can assign a different IP address to the gateway, using one of the following methods:

- BootP (refer to Section 3.2.1 on page 14)
- Command Line Interface (CLI) using the RS-232 interface (refer to Section 3.2.2 on page 15)
- Gateway's HTTP-based Embedded Web Server (refer to Section 3.2.3 on page 16)
- DHCP (refer to the gateway's User's Manual)

Table 3-1: Gateway Default Networking Parameters

Network Parameter	Default Value
IP address	10.1.10.11
Default subnet mask	255.255.0.0
Default gateway IP address	0.0.0.0

You can use the hardware reset button to restore the gateway's networking parameters to their factory default values (refer to Section 6.2 on page 40).

3.2.1 Assigning an IP Address using BootP

You can use AudioCodes' Bootstrap Protocol (BootP) or any third-party BootP application to assign an IP address to the gateway.

➢ To assign an IP address using BootP, take these 3 steps:

1. Open the BootP application.
2. Add a new client by performing the following (for AudioCodes BootP Server application):
   a. From the Services menu, choose **Clients**, the Client Configuration dialog box appears.
   b. Add a client configuration for the gateway that you want to initialize, by clicking the **Add New Client** button.
   c. Enter the necessary fields (i.e., gateway's MAC address and desired IP address), and then click **OK**.

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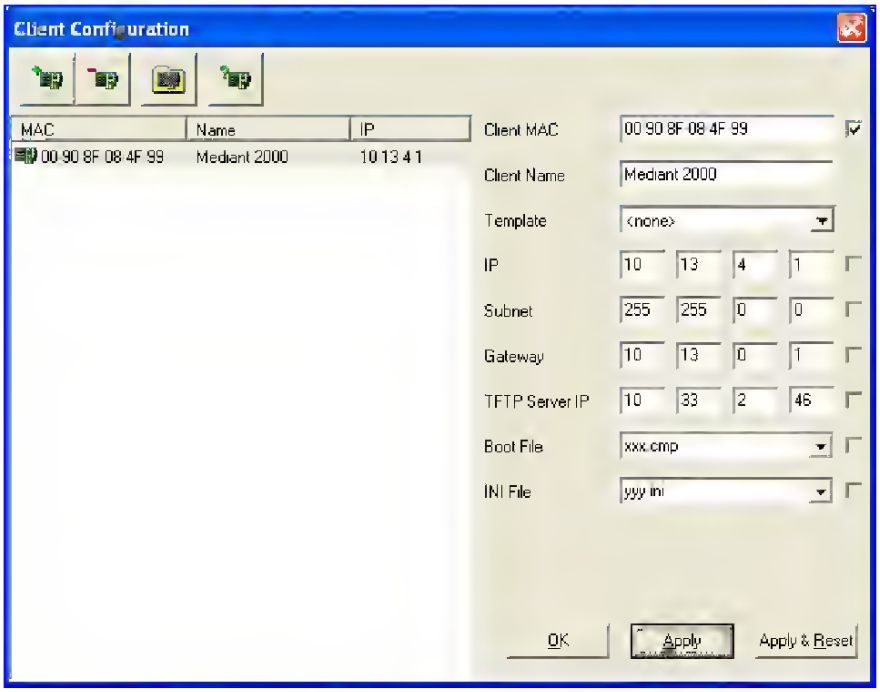
Table 3-2: Adding a BootP Client



3. Use the gateway's chassis reset button to physically reset the gateway, causing it to use BootP, the gateway changes its network parameters to the values provided by BootP

3.2.2 Assigning an IP Address Using CLI via RS-232

You can assign an IP address to the gateway using CLI by establishing a serial connection between your PC and the gateway

➢ To assign an IP address using CLI via RS-232, take these 5 steps:

1. Connect the gateway's RS-232 port (labeled **RS-232**), located on the rear panel, to your PC
2. Use serial communication software (e.g., HyperTerminal™) to connect to the gateway. Set your serial communication software to the following communications port settings: Baud Rate: 115,200 bps, Data bits: 8, Parity: None, Stop bits: 1, Flow control: None
3. At the CLI prompt, type **conf**, and then press <Enter>, the configuration folder is accessed
4. To check the current network parameters, at the prompt, type **GCP IP**, and then press <Enter>; the current network settings are displayed
5. Change the network settings by typing the following:

   SCP IP [ip_address] [subnet_mask] [default_gateway]

   (e.g., **SCP IP 10.13.77.7 255.255.0.0 10.13.0.1**)

   Note: This command requires you to enter all three network parameters (each separated by a space)

The new settings take effect on-the-fly and connectivity is active at the new IP address. To save the configuration, at the prompt, type **SAR**, and then press <Enter>, the gateway restarts with the new network settings

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3.2.3 Assigning an IP Address using HTTP

You can assign an IP address to the gateway using the gateway's Embedded Web Server (based on HTTP), by connecting the gateway to a PC with a direct, local Ethernet connection.

➢ **To assign an IP address to the gateway using HTTP, take these 5 steps:**

1. Connect your PC to the gateway by performing the following:
   a. Disconnect the gateway from the network and reconnect it to your PC using one of the following methods:
      - Use a standard Ethernet cable to connect the network interface on your PC to a port on a network hub or switch. Use a second standard Ethernet cable to connect the gateway to another port on the same network hub or switch.
      - Use an Ethernet cross-over cable to directly connect the network interface on your PC to the gateway.
   b. Change your PC's IP address and subnet mask to correspond with the gateway's factory default IP address and subnet mask (listed in Table 3-1).
2. Access the gateway's Embedded Web Server (refer to Section 3.1.1 on page 10), the 'Quick Setup' screen is displayed (**Quick Setup** menu), as shown in the figure below.

Figure 3-6: Quick Setup Screen

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3. Configure the gateway's 'IP Address', 'Subnet Mask', and 'Default Gateway IP Address' fields to correspond with your network IP settings of Office Communications Server 2007.
4. Click the **Reset** button, and then at the prompt, click **OK**, the gateway applies the changes and restarts. This takes approximately three minutes to complete, after which the **Power** and **Ready** LEDs on the front panel are lit green.
5. Connect the gateway to the network by performing the following:
   a. Disconnect your PC from the gateway or from the hub / switch (depending on the connection method you used in Step 1).
   c. Reconnect the gateway and your PC (if necessary) to the network.
   b. Restore your PC's original IP address and subnet mask. Re-access the gateway using the Embedded Web Server with its newly assigned IP address.

Tip: Record the IP address and subnet mask you assigned to the gateway. Do the same when defining new username or password. If the Embedded Web Server is unavailable (for example, if you've lost your username and password), use the BootP/TFTP configuration utility to access the device, "reflash" the load and reset the password (refer to the gateway's User's Manual for detailed information on using a BootP/TFTP configuration utility to access the device).

3.3 Advanced Configuration

Once you have IP connectivity to the gateway (as described in Section 3.2 on page 14), you can configure the advanced gateway parameters for interoperability with the deployed PBX / PSTN interfaces and Office Communications Server 2007.

The advanced parameters can be configured using one of the following methods:

- Pre-configured configuration file (referred to as the *ini* file) uploaded to the gateway (refer to Section 3.3.1 on page 17)
- Gateway's Embedded Web Server management tool (refer to Section 3.3.2 on page 20)

Tip: Once the gateway is configured, backup your settings by saving a copy of the VoIP gateway configuration file (*ini*) to a directory on your PC (refer to Section 4 on page 37). This saved file can later be used, if necessary, to restore previous configuration settings (refer to Section 3.3.1.2 on page 18).

3.3.1 Using the *ini* File

The *ini* file is a text-based file that contains the gateway's configurable parameters required for operating in the specific network environment. To configure the gateway using the *ini* file, the following steps must be performed:

- Obtain the *ini* file from AudioCodes' Web site (refer to Section 3.3.1.1 on page 17)
- Upload the *ini* file to the gateway (refer to Section 3.3.1.2 on page 18)
- Modify parameters specific to the deployment site (refer to Section 3.3.1.3 on page 19)

3.3.1.1 Obtaining the *ini* File from the Web

Before you can load the required *ini* file to your gateway, you need to obtain the file from AudioCodes' Web site. From this site, you can also download the latest User's Manuals, training documentation, and firmware.

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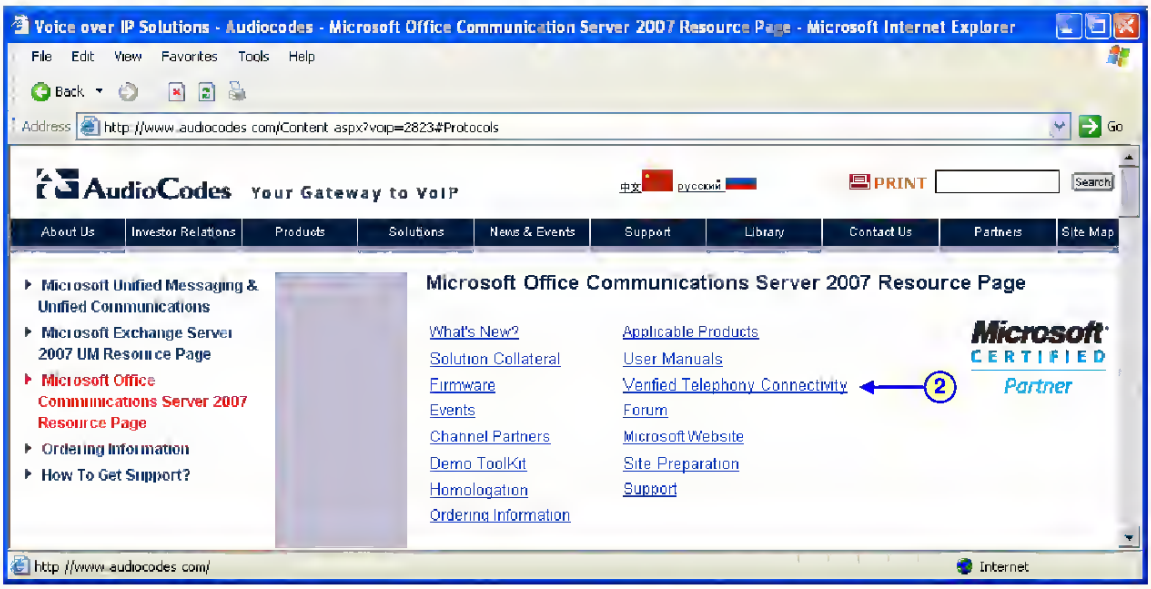
Notes:

- Ensure that you download the correct *ini* file so that your gateway is configured correctly for interoperability with the deployed PBX / PSTN
- The gateway configuration for the PBX-Sends-Digits option is similar to the One-to-Many option. Therefore, for the PBX-Sends-Digits option, download the *ini* file for the One-to-Many option.

➢ To download the *ini* file to your PC, take these 4 steps:

1. Open your Web browser, and then in the URL address field, enter the following address: http://www.audiocodes.com/Content.aspx?voip=2823, the 'Microsoft Office Communications Server 2007 Resource Page' page opens, as shown below:

Figure 3-7: Microsoft Office Communications Server 2007 Resource Page



2. Navigate to the 'Verified Telephony Connectivity' table (which contains *ini* files per tested PBX), by either clicking **Verified Telephony Connectivity** or scrolling down to the table.
3. In the 'Download' column, click the ZIP file icon corresponding to the required PBX / PSTN protocol; the 'File Download' message box appears.
4. Click the **Save** button, navigate to the folder on your PC to where you want to download the file, and then click **Save**; the file is downloaded to the folder and when complete, the 'Download Complete' message box appears.

3.3.1.2 Uploading the *ini* File to the Gateway

Once you have downloaded the correct *ini* file from the Web, you need to upload it to the gateway's non-volatile memory using the gateway's Embedded Web Server.

➢ To upload the *ini* file to the gateway, take these 6 steps:

1. Unzip the downloaded ZIP file that you downloaded in the previous section. An *ini* file (with the file extension *.ini) is extracted along with any relevant files.
2. Login to the gateway's Embedded Web Server (refer to Section 3.1.1 on page 10).
3. Open the 'Configuration File' screen (**Advanced Configuration** menu > **Configuration File**).

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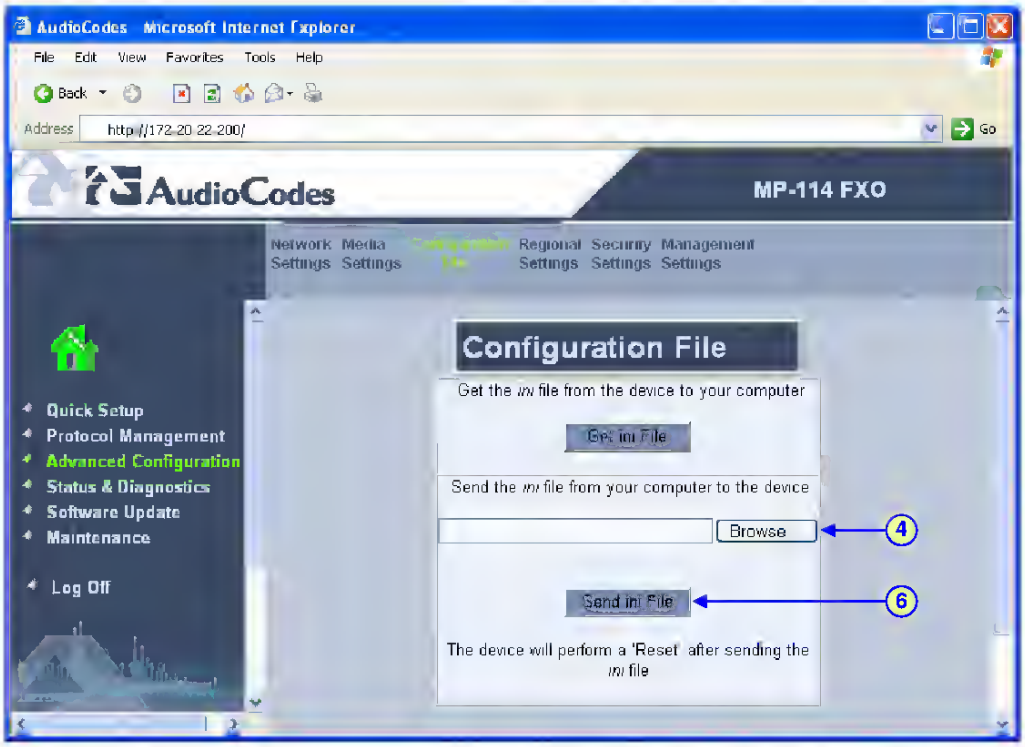
Figure 3-8: Configuration File Screen



4. Click the **Browse** button, and then navigate to the folder that contains the *ini* file you want to upload.
5. Select the file, and then click the **Open** button; the name and path of the file appear in the field beside the **Browse** button.
6. Click the **Send *ini* File** button, and then at the prompt, click **OK**; the gateway automatically resets (from the version stored on the flash memory).

3.3.1.3 Modifying Parameters Specific to Site Deployment

Typically, for interoperability with the deployed PBX / PSTN interface and Office Communications Server 2007, it's sufficient that you use the *ini* file obtained from AudioCodes' Web page. This file contains all the required parameter settings for the gateway to seamlessly operate in this application setup. However, due to specificity of site deployment, the following configurations may need to be configured or modified:

- IP address of Mediation Server (refer to Section 3.3.2.1 on page 20)
- Number of FXO ports connected to the PBX (refer to Section 3.3.2.2 on page 22)
- Voice coders (refer to Section 3.3.2.3 on page 23)
- Numbers translation (normalization) from/to E.164 format using Manipulation tables for PBX / PSTN connectivity - depends on deployment dial plan (refer to Section 3.3.2.4 on page 23)
- Answer Supervision (refer to Section 3.3.2.5 on page 28)
- Automatic Dialing (refer to Section 3.3.2.6 on page 29)
- Caller ID (refer to Section 3.3.2.7 on page 30)

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3.3.2 Using the Embedded Web Server

You can use the gateway's user-friendly, Web interface (described in Section 3.1 on page 10) to configure the advanced parameters listed below.

- IP address of Mediation Server (refer to Section 3.3.2.1 on page 20)
- Number of FXO ports connected to the PBX (refer to Section 3.3.2.2 on page 22)
- Voice coders (refer to Section 3.3.2.3 on page 23)
- Numbers translation (normalization) from/to E.164 format using Manipulation tables for PBX / PSTN connectivity -- depends on deployment dial plan (refer to Section 3.3.2.4 on page 23)
- Answer Supervision (refer to Section 3.3.2.5 on page 28)
- Automatic Dialing (refer to Section 3.3.2.6 on page 29)
- Caller ID (refer to Section 3.3.2.7 on page 30)
- SIP transport type (refer to Section 3.3.2.8 on page 31)
- Disconnect on broken connection (refer to Section 3.3.2.9 on page 32)
- DTMF Payload Type (refer to Section 3.3.2.10 on page 33)
- One-Stage Dialing (refer to Section 3.3.2.11 on page 34)
- Channel Select Mode (refer to Section 3.3.2.12 on page 35)
- RTCP Attribute (refer to Section 3.3.2.13 on page 36)

3.3.2.1 Defining Mediation Server's IP Address

As the IP address of Mediation Server (through which the gateway communicates with Office Communications Server 2007) is specific to deployment, you must define this parameter. The gateway forwards all telephone calls (PBX or PSTN) to this IP address (i.e., to Mediation Server).

The IP address is defined in the gateway's Embedded Web Server as the proxy server's IP address. In other words, Mediation Server acts as a proxy server (without registration).

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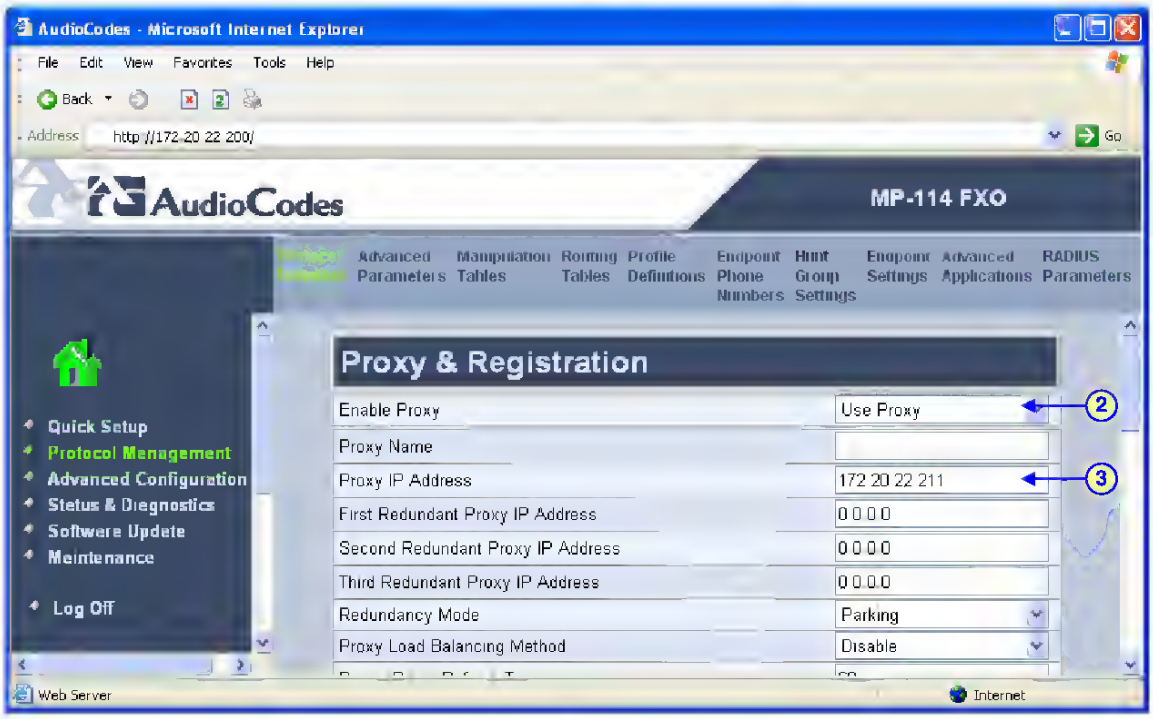
➢ To define Mediation Server's IP address, take these 4 steps:

1. Open the 'Proxy & Registration' screen (**Protocol Management** menu > **Protocol Definition** > **Proxy & Registration**)

Figure 3-9: Proxy & Registration Screen



2. From the 'Enable Proxy' drop-down list, select 'Use Proxy'. This allows Mediation Server to act as a proxy server.
3. In the 'Proxy IP Address' field, enter the IP address (or fully qualified domain name -- FQDN) of Mediation Server.

   Note: When using FQDN, ensure that you define the DNS server's IP address (in the 'IP Settings' screen -- **Advanced Configuration** > **Network Settings** > **IP Settings**) or Mediation Server's domain name and corresponding IP address (in the 'Internal DNS Table' screen -- **Protocol Management** > **Routing Tables** > **Internal DNS Table**).
4. Click **Submit** to save your settings.

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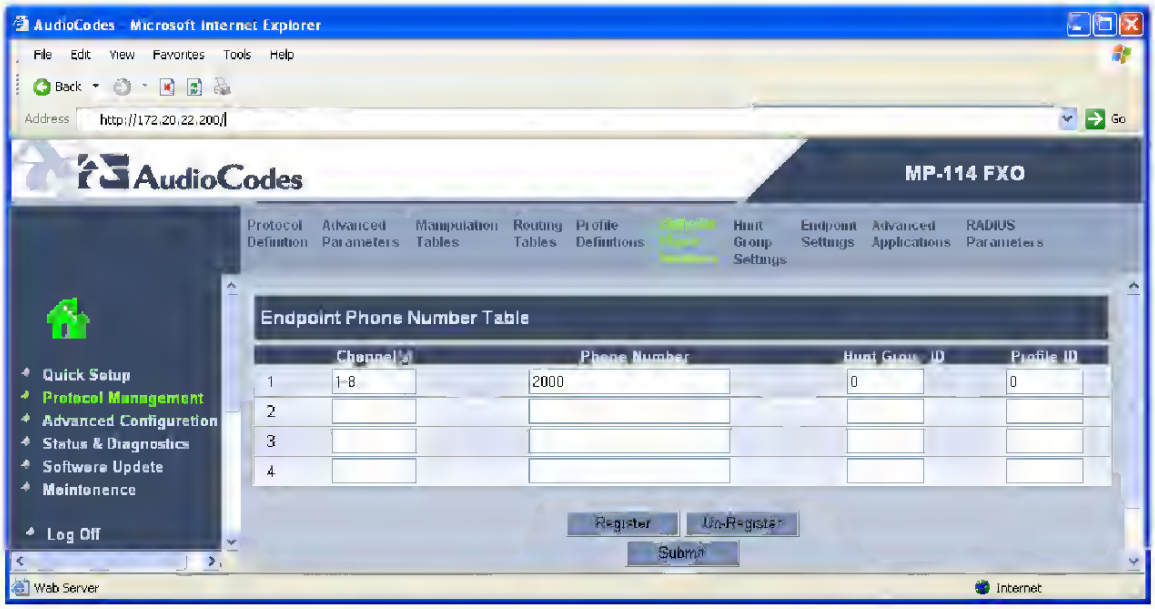
3.3.2.2 Defining Number of FXO Ports Connected to PBX

The gateway's FXO port (located on the rear panel), interfaces with the PBX. By default, all the gateway's FXO ports (eight for the MP-118, four for the MP-114) are enabled. However, if for any reason the ports (*endpoints*) are disabled, you can enable them by performing the procedure below.

➢ To enable the gateway's FXO ports, take these 4 steps:

1. Open the 'Endpoint Phone Number Table' screen (**Protocol Management** menu > **Endpoint Phone Numbers**)

Figure 3-10: Endpoint Phone Number Table Screen



2. In the 'Channel(s)' field, enter the gateway's port number that is connected to the PBX. You can enter all the ports, by simply entering a range in the first table row (e.g., '1-8' for all eight ports of the MP-118).
3. In the 'Phone Number' field, enter any logical telephone number (e.g., '2000') for these channels.
4. Click **Submit** to save your settings.

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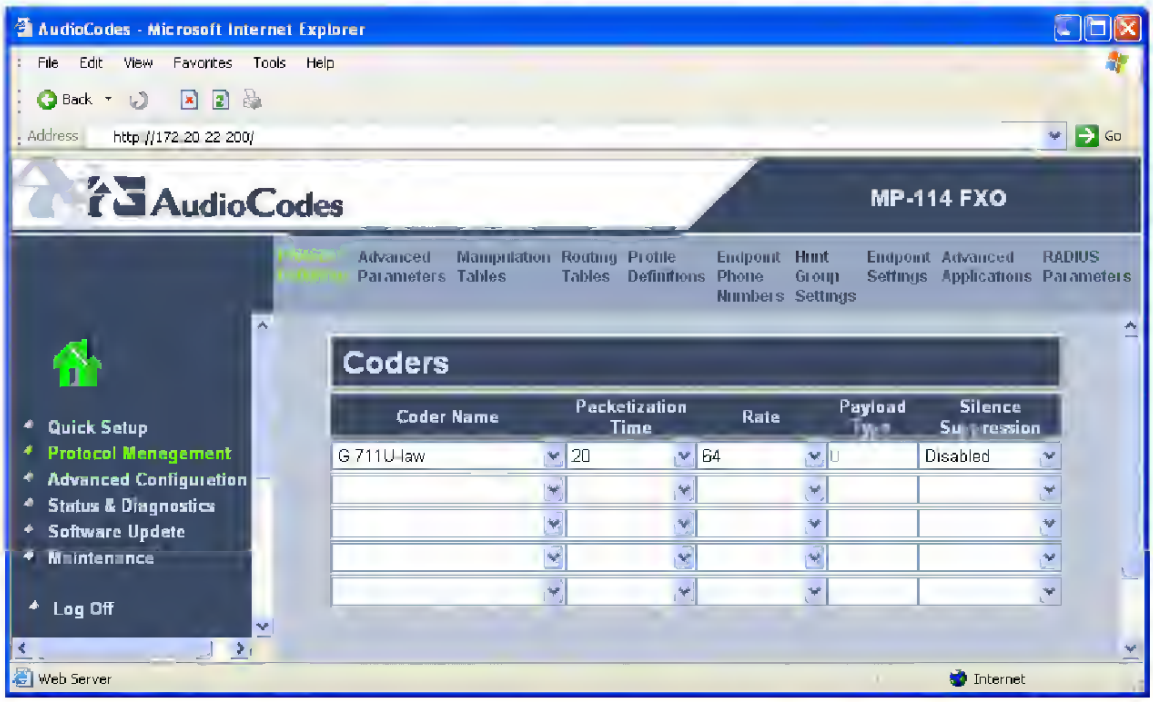
3.3.2.3 Defining Voice Coders

The gateway communicates with Mediation Server using either the G.711 A-law or G.711 µ-law coder. You can change the default coder using the gateway's Embedded Web Server.

➢ To define the coder, take these 3 steps:

1. Open the 'Coders' screen (**Protocol Management** menu > **Protocol Definition** > **Coders**)

Figure 3-11: Coders Screen



2. From the 'Coder Name' drop-down list, select the required coder.
3. Reset the gateway by clicking the **Maintenance** button on the main menu bar. Select the 'BURN' option to burn the configuration to flash memory, and then click the **Reset** button.

3.3.2.4 Translating Numbers From / To E.164 Using Manipulation Tables for PBX/PSTN Connectivity

The Embedded Web Server's Manipulation tables provide the ability to translate (*normalize*) numbers dialed in standard E.164 format to various formats and vice versa. Manipulation is necessary for your dial plan as Office Communications Server 2007 uses the standard E.164 format while your PBX or PSTN implements other number formats for dialing.

Due to Office Communications Server 2007normalization rules, the gateway may need to perform number manipulation for outbound calls (i.e., calls received from OCS client via Office Communications Server 2007) and inbound calls (i.e., calls destined for OCS client). If the gateway is connected to a PBX or directly to the PSTN, the gateway may need to perform number manipulations for the called and/or calling number to match the PBX or PSTN interfaces dialing rules or to match Office Communications Server 2007 E.164 format.

Manipulation number configuration examples are provided for inbound and outbound calls in Section 3.3.2.4.2 on page 26.

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➢ To configure the Number Manipulation tables, take these 4 steps:

1. Open the Number Manipulation screen you want to configure (**Protocol Management** menu > **Manipulation Tables**), the relevant Manipulation table screen is displayed. The figure below shows the 'Source Phone Number Manipulation Table for Tel→IP calls'

Figure 3-12: Source Phone Number Manipulation Table for Tel→IP Calls

	Dest. Prefix	Source Prefix	Num of Stripped Digits	Prefix (Suffix) to Add	Number of Digits to Leave	Presentation
1	03	201	0	972		Allowed
2		1001	4	5(23)		Restricted
3		123451001#	0	(8)	4	Not Configured
4		[30-40]xx	(1)	2		Not Configured
5	[6,7,8]	2001	5	3		Not Configured
6						Not Configured

2. In the 'Table Index' drop-down list, select the range of entries that you want to edit (up to 20 entries can be configured for Source Number Manipulation and 50 entries for Destination Number Manipulation).

   The figure above exemplifies the use of manipulation rules performed on source telephone numbers (for a description on dialing plan notations, refer to Section 3 3 2 4 1 in page 25)

   - When the destination number is 035000 and source number is 20155, the source number is changed to 97220155 (according to Rule 1)
   - When source number equals 1001876, it is changed to 587623 (according to Rule 2)
   - Source number 1234510012001 is changed to 20018 (according to Rule 3)
   - Source number 3122 is changed to 2312 (according to Rule 4)
3. Configure the Number Manipulation table according to Table 3-3
4. Click the **Submit** button to save your changes

Notes:

- The manipulation rules are applied to any incoming call whose 1) destination number prefix matches prefix defined in the 'Destination Number' field, 2) source number prefix matches the prefix defined in the 'Source Prefix' field, and 3) source IP address matches IP address defined in the 'Source IP' field (if applicable)
- The number manipulation can be performed using a combination of each of the above criteria, or using each criterion independently
- The manipulation rules are executed in the following order
  1) Number of stripped digits
  2) Number of digits to leave
  3) Prefix / suffix to add

Table 3-3: Number Manipulation Parameters Description

Parameter	Description
Destination Prefix	Destination (called) telephone number prefix. An asterisk (*) represents any number.
Source Prefix	Source (caller) telephone number prefix. An asterisk (*) represents any number.

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Table 3-3: Number Manipulation Parameters Description

Parameter	Description
Source IP (Applicable only to the 'Destination Phone Number Manipulation Table for IP to Tel')	Source IP address of the call (obtained from the Contact header in the INVITE message) **Note:** The source IP address can include the 'x' wildcard to represent single digits. For example, 10.8.8.xx represents all the addresses between 10.8.8.10 to 10.8.8.99. In addition, the asterisk (*) wildcard represents any number between 0 and 255, e.g., 10.8.8.* represents all addresses between 10.8.8.0 and 10.8.8.255.
Num of Stripped Digits	▪ Enter the number of digits that you want to remove from the left of the telephone number prefix. For example, if you enter 3 and the phone number is 5551234, the new phone number is 1234. ▪ Enter the number of digits in parenthesis ( ) that you want to remove from the right of the telephone number prefix. **Note:** A combination of the two options is allowed, e.g., 2(3)
Prefix / Suffix to Add	▪ Add to prefix. Enter the number / string you want to add to the front of the phone number. For example, if you enter 9 and the phone number is 1234, the new number is 91234. ▪ Add to suffix. Enter the number / string in parenthesis you want to add to the end of the phone. For example, if you enter (00) and the phone number is 1234, the new number is 123400. **Note:** You can enter a prefix and a suffix in the same field, e.g., 9(00)
Number of Digits to Leave	Enter the number of digits that you want to leave from the right.
Presentation	Select 'Allowed' to send Caller ID information when a call is made using these destination / source prefixes. Select 'Restricted' if you want to restrict Caller ID information for these prefixes.

3.3.2.4.1 Dialing Plan Notation

The dialing notation applies to numbers entered in the 'Destination Prefix' and 'Source Prefix' fields of the Manipulation tables to represent multiple numbers.

Table 3-4: Dialing Plan Notations

Notation	Description	Example
[n-m]	Represents a range of numbers **Note:** range of letters is not supported.	[5551200-5551300]# Represents all numbers from 5551200 to 5551300. 123[100-200]# Represents all numbers from 123100 to 123200.
[n,m]	Represents multiple numbers. **Note:** This notation only supports single-digit numbers.	[2,3,4]xxx# Represents four-digit numbers that start with 2, 3 or 4.
x	Represents any single digit.	54324. Represents any number that starts with 54324.
Pound sign (#) at the end of a number	Represents the end of a number.	54324xx#. Represents a 7 digit number that starts with 54324.
A single asterisk (*)	Represents any number.	*. Represents any number.

The gateway matches the rules starting at the top of the table (i.e., top rules take precedence over lower rules). For this reason, enter more specific rules above more generic rules. For example, if you enter 551 in entry 1 and 55 in entry 2, the gateway applies rule 1 to numbers that start with 551 and applies rule 2 to numbers that start with 550, 552, 553, and so on (skipping 551). However if you enter 55 in entry 1 and 551 in entry 2, the gateway applies rule 1 to all numbers that start with 55 including numbers that start with 551.

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3.3.2.4.2 Number Normalization Examples

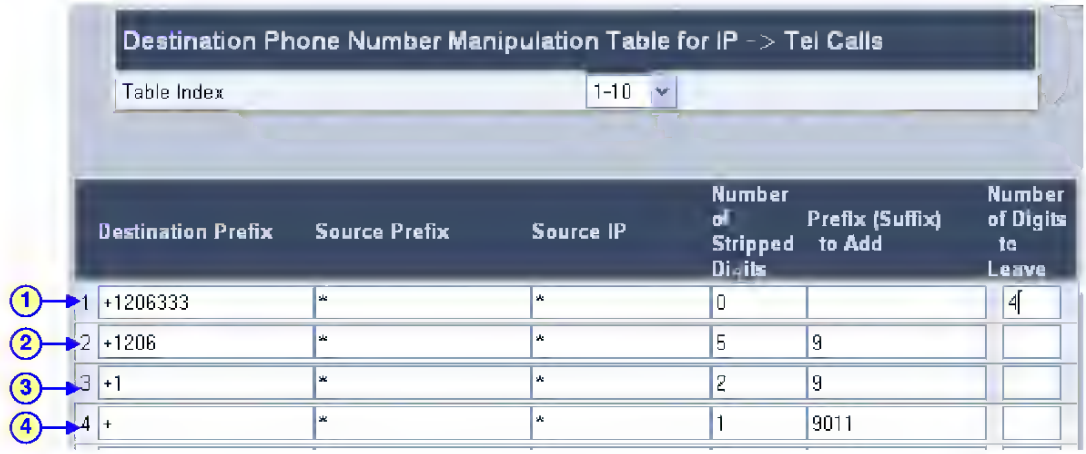
Two examples are provided below for number normalization. The examples are based on the following assumptions: a PBX with prefix (local) number 333 and a 4-digit extension number that begins with the digit 1 (i.e., 1xxx), National area code 206, International code 1.

- **Modifying E.164 Numbers to PBX Format for Outbound Calls:** Outbound calls refer to calls made by Office Communications Server 2007 users (OCS clients) connected through IP to the Office Communications Server 2007.

  1. **Local calls within the PBX:** The caller dials only the last four digits (e.g., 1212). Office Communications Server 2007 translates (normalizes) the phone number into an E.164 number format +12063331212 (where *+1* is the country code, *206* the local area code, and *333* the PBX prefix number). The gateway's Manipulation table is configured to send only the last four digits to the PBX (i.e., 1212).
  2. **National calls to the same area code:** The caller dials 9 for an external line and then dials a 7-digit telephone number (e.g., 9-555-4321). Office Communications Server 2007 translates (normalizes) the phone number into an E.164 number format +12065554321 (where *+1* is the country code, *206* the local area code, *5554321* the phone number). The gateway's Manipulation table is configured to remove (strip) the first five digits and add 9 as a prefix to the remaining number. Therefore, the gateway sends the number 95554321 to the PBX, and then the PBX sends the number 5554321 to the PSTN.
  3. **National calls to a different area code:** The caller dials 9 for an external line, the out-of-area code, and then a 7-digit telephone number (e.g., 9-503-331-1425). Office Communications Server 2007 translates (normalizes) the phone number into an E.164 number format +15033311425 (where *+1* is the international code, *503* the out-of area code, *3311425* the phone number). The gateway's Manipulation table is configured to remove (strip) the first two digits (i.e., *+1*), add then add 9 as a prefix to the remaining number. Therefore, the gateway sends the number 95033311425 to the PBX and then the PBX sends the number 5033311425 to the PSTN.
  4. **Making international calls:** The caller dials 9 for an external line, the access code for international calls (e.g., 011 for the US), the country code (e.g., +44 for the UK), the area code (e.g., 1483), and then a 6-digit telephone number (e.g., 829827). Office Communications Server 2007 translates (normalizes) the phone number into an E.164 number format +441483829827 (where *+44* is the country code, *1483* the area code, *829827* the phone number). The gateway's Manipulation table is configured to remove the first digit (e.g., +), and add the external line digit (e.g., 9) and the access code for international calls (e.g., 011 for the US) as the prefix. Therefore, the gateway sends the number 9011441483829827 to the PBX and the PBX, in turn, sends the number 011441483829827 to the PSTN.

The configuration of the above scenarios is shown in the figure below:

Figure 3-13: Phone Number Manipulation Table for IP→Tel Calls



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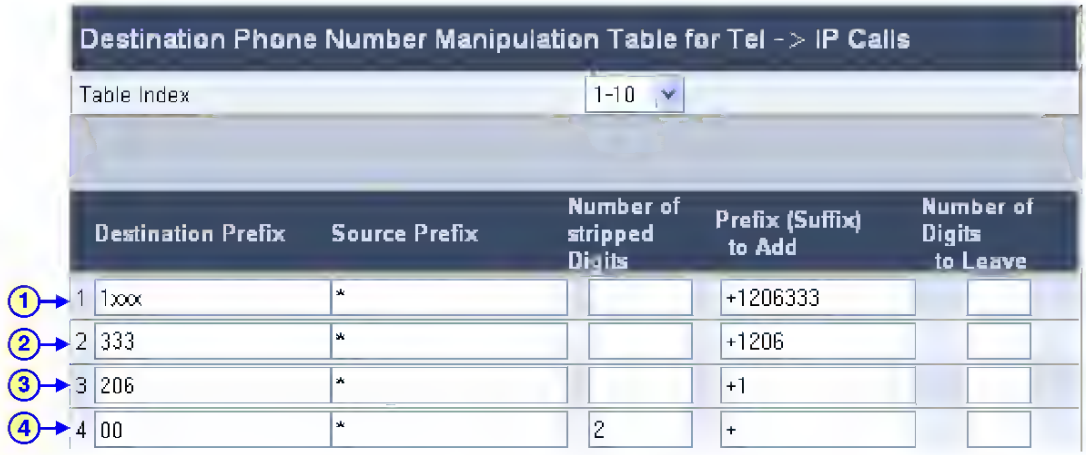
- **Modifying PBX, Local, and National Calls to E.164 Format for Inbound Calls:** Inbound calls refer to calls received by OCS clients connected through IP to the Office Communications Server 2007.

  1. **Local calls from the PBX or PSTN:** the PBX user only dials a 4-digit extension number of the OCS client (e.g., 1220). The gateway's Manipulation table is configured to normalize the number into E.164 format and adds the prefix +1206333 to the extension number. Therefore, the gateway sends the number +12063331220 to Office Communications Server 2007, which relays the call to the OCS client.
  2. **National calls with the same area code**: the PSTN user dials a 7-digit phone number (e.g., 333-1220), which is received by the gateway. The gateway's Manipulation table is configured to normalize the number into E.164 format and adds the prefix +1206 to the number. Therefore, the gateway sends the number +12063331220 to Office Communications Server 2007, which relays the call to the OCS client.
  3. **National calls from a different area code:** the PSTN user dials the national area code and then a 7-digit phone number (e.g., 206-333-1220), which is received by the gateway. The gateway's Manipulation table is configured to normalize the number into E.164 format and adds the prefix +1 to the number. Therefore, the gateway sends the number +12063331220 to Office Communications Server 2007, which relays the call to the OCS client. **Note:** Whether the area code is received by the gateway depends on the country's PSTN numbering rules.
  4. **International calls:** The PSTN international (overseas) caller dials the international access and country code (e.g., 001 for the US), the national area code, and then a 7-digit phone number (e.g., 206-333-1220), which is received by the gateway. The gateway's Manipulation table is configured to normalize the number into E.164 format by removing the first two digits (e.g., 00) and adding the prefix plus sign (+). Therefore, the gateway sends the number +12063331220 to Office Communications Server 2007, which relays the call to the OCS client. **Note:** Whether the international and country codes are received by the gateway depends on the country's PSTN numbering rules.

  The configuration of the above scenarios is shown in the figure below:

Figure 3-14: Phone Number Manipulation Table for Tel→IP Calls



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3.3.2.5 Defining Answer Supervision

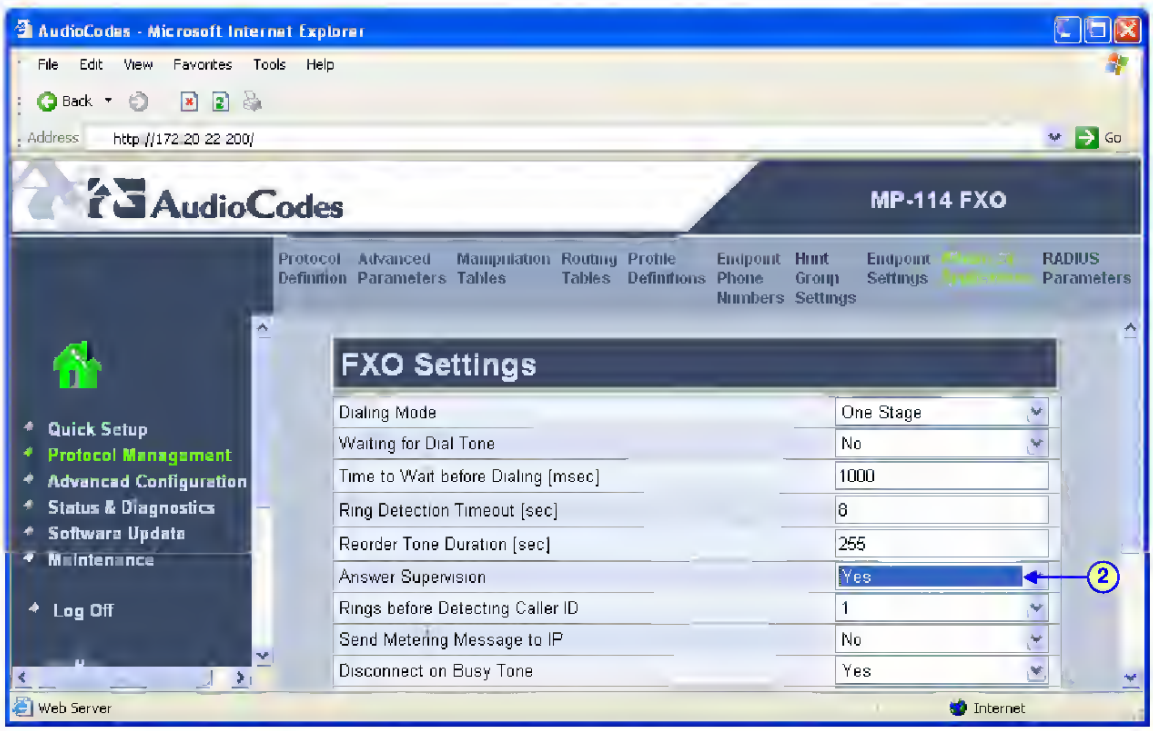
When there is a call from Office Communications Server 2007 to the gateway, the gateway behavior depends on whether or not the Answer Supervision feature is enabled. If Answer Supervision is enabled, the gateway detects (by speech detection) when the called party answers the call (i.e., if the called party says "hello" or any other word), and only then sends a 200 OK + SDP SIP message.

When implementing the **Ring Additional Number** feature (and the additional number is a PBX/PSTN phone) of Office Communications Server 2007, Answer Supervision must be enabled. This ensures that the gateway sends the 200 OK + SDP only if the PBX/PSTN phone answers the call and starts talking.

➢ To define the Answer Supervision, take these 3 steps:

1. Open the 'FXO Setting' screen (**Protocol Management** menu > **Advanced Applications** > **FXO Settings**).

Figure 3-15: FXO Settings Screen



2. From the 'Answer Supervision' drop-down list, select 'Yes'.
3. Click the **Submit** button to save your changes.

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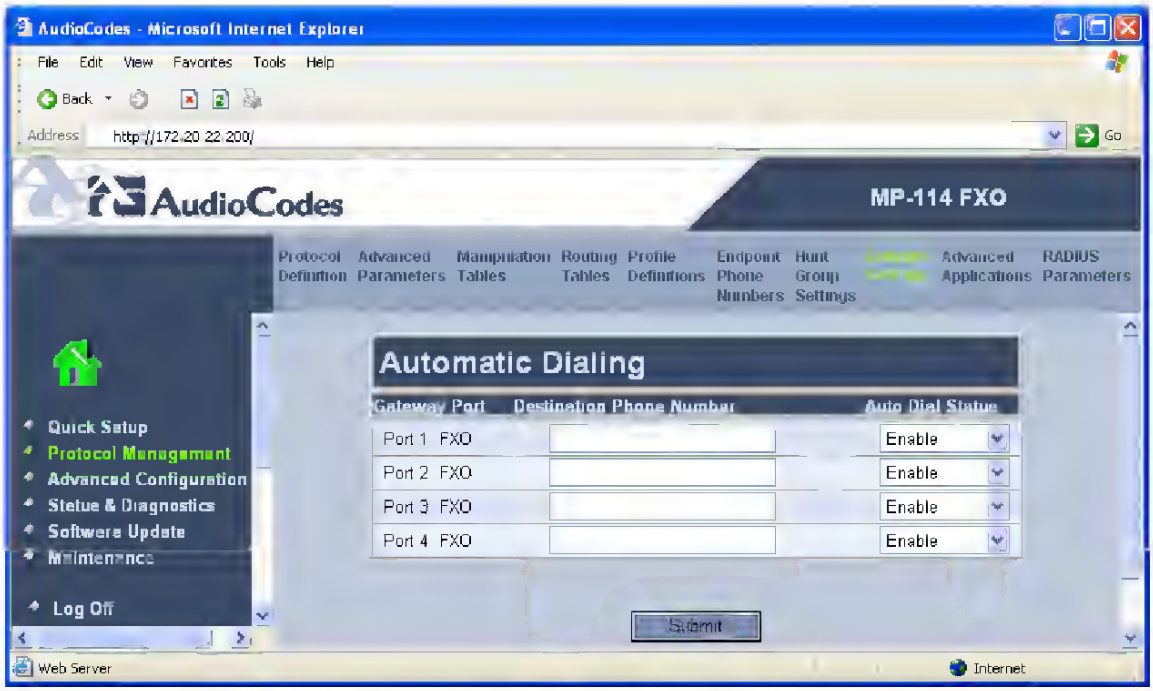
3.3.2.6 Defining Automatic Dialing

When using the One-to-One option, each PBX line that is connected to an FXO port serves one OCS client. You must define automatic dialing for each FXO port connected to the OCS client.

➢ To define Automatic Dialing, take these 4 steps:

1. Open the 'Automatic Dialing' screen (**Protocol Management** menu > **Endpoint Settings** > **Automatic Dialing**).

Figure 3-16: Automatic Dialing Screen



2. In the 'Destination Phone Number' field, enter the OCS client's phone number that this FXO port serves.
3. From the 'Auto Dial Status' drop-down list, select 'Enable'.
4. Click the **Submit** button to save your changes.

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3.3.2.7 Enabling Caller ID

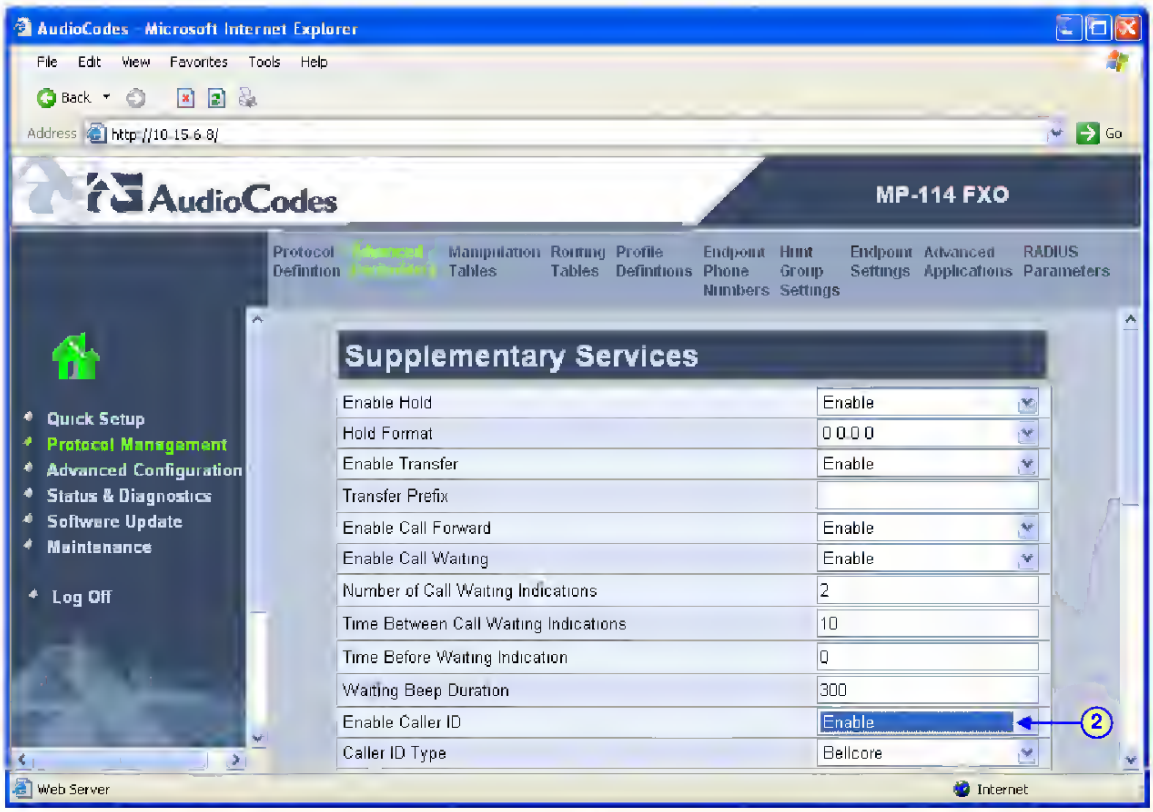
If the Caller ID service is enabled, the gateway detects the Caller ID signal from the Tel side (PBX / PSTN) and sends it to Mediation Server over IP in the SIP INVITE message. Therefore, to allow the OCS clients to receive caller ID, enable the Caller ID service at the gateway.

Note: If the deployed PBX does not support Caller ID (and thus, the gateway cannot provide Caller ID), it's recommended to disable the Caller ID service in order to prevent a delay in call establishment.

➢ **To enable Caller ID, take these 3 steps:**

1. Open the 'Supplementary Services' screen (**Protocol Management** menu > **Advanced Parameters** > **Supplementary Services**).

Figure 3-17: Supplementary Services Screen



2. From the 'Enable Caller ID' drop-down list, select 'Enable'.
3. Click the **Submit** button to save your changes.

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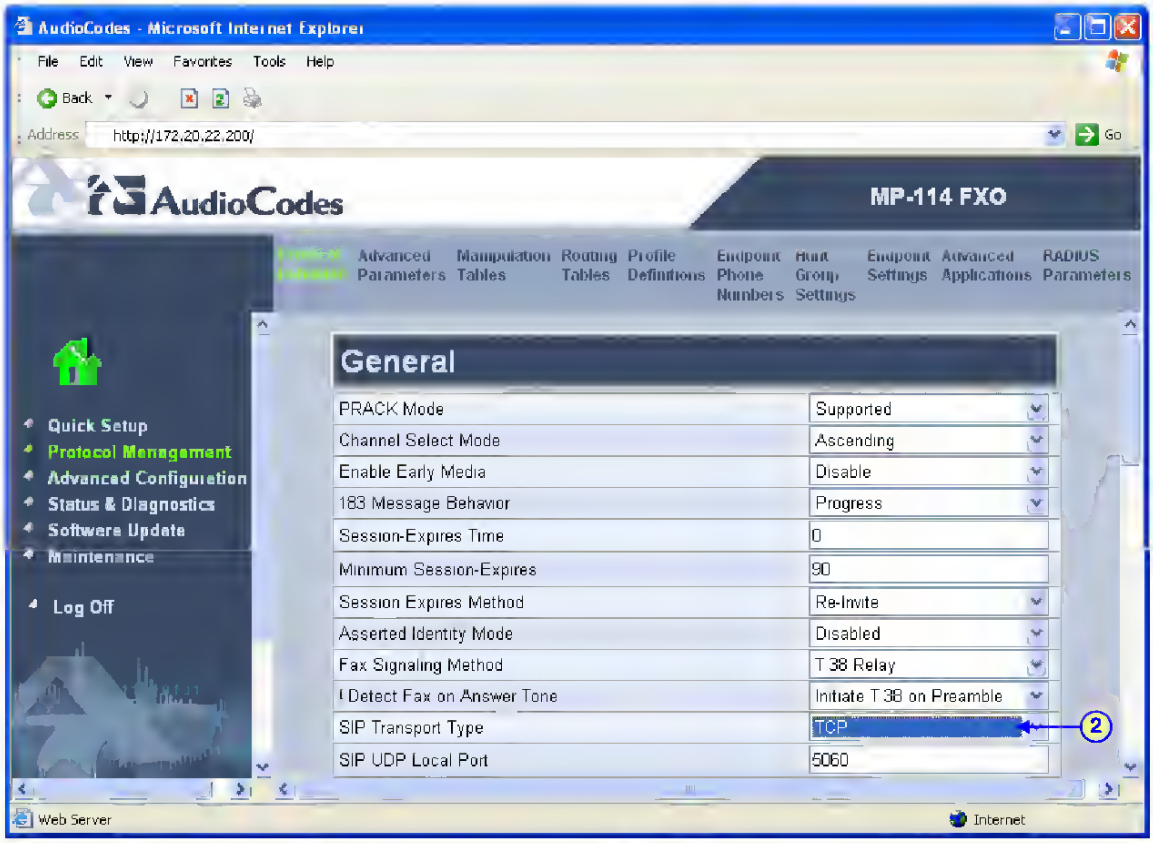
3.3.2.8 Defining SIP Transport Type

The procedure below describes how to configure the SIP transport type. You may need to change the gateway's SIP transport type from UDP (default) to TCP, since this is the transport type used by Mediation Server.

➢ **To define the SIP transport type, take these 3 steps:**

1. Open the 'General Parameters' screen (**Protocol Management** menu > **Protocol Definition** > **General Parameters**)

Figure 3-18: General Parameters Screen



2. From the 'SIP Transport Type' drop-down list, select 'TCP'.
3. Click the **Submit** button to save your changes.

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3.3.2.9 Defining Disconnect on Broken Connection

The procedure below describes how to configure the Disconnect on Broken Connection parameter. Since Mediation Server uses Silence Suppression, it may not send Real-time Transport Protocol (RTP) packets during the detection of silence in the call. To avoid call disconnection by the gateway (due to RTP packets not being transmitted), you may need to disable the parameter 'Disconnect on Broken Connection'.

➢ **To define Disconnect on Broken Connection, take these 3 steps:**

1. Open the 'General Parameters' screen (**Protocol Management** menu > **Advanced Parameters** > **General Parameters**)

Figure 3-19: General Parameters Screen

2. From the 'Disconnect on Broken Connection' drop-down list, select 'No'.
3. Click the **Submit** button to save your changes.

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3.3.2.10 Defining DTMF Payload Type

The procedure below describes how to define the dual-tone multi-frequency (DTMF) payload type The DTMF payload type used by Mediation Server is 101 You may need to set this value at the gateway as well

➢ **To define the DTMF payload type, take these 3 steps:**

1. Open the 'DTMF & Dialing' screen (**Protocol Management** menu > **Protocol Definition** > **DTMF & Dialing**)

Figure 3-20: DTMF & Dialing Screen

2. Set the 'RFC 2833 Payload Type' field to 101
3. Click the **Submit** button to save your changes

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3.3.2.11 Enabling One-Stage Dialing

One-stage dialing is when the gateway receives an IP-to-Tel call, off-hooks the PBX line connected to the PBX / PSTN telephone, and then immediately dials the destination telephone number. In other words, the OCS client (i.e., IP caller) doesn't dial the PSTN number upon hearing a dial tone.

➢ To enable one-stage dialing, take these 3 steps:

1. Open the 'FXO Settings' screen (**Protocol Management** menu > **Advanced Applications** > **FXO Settings**), the 'FXO Settings' screen is displayed.

Figure 3-21: FXO Settings Screen

2. From the 'Dialing Mode' drop-down list, select 'One Stage'
3. Click the **Submit** button to save your changes

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3.3.2.12 Defining Channel Select Mode

Channel select modes determine the method in which new IP-to-Tel calls (i e , calls from OCS clients to PBX/PSTN phones) are assigned to gateway ports (i e , channels) It is recommended to select the Cyclic Ascending mode, which selects the next available channel in an ascending cycle order

➢ **To define the Channel Select mode, take these 3 steps:**

1. Open the 'General' screen (**Protocol Management** menu > **Protocol Definition** > **General Parameters**)

Figure 3-22: General Screen

2. From the 'Channel Select Mode' drop-down list, select 'Cyclic Ascending'.
3. Click the **Submit** button to save your changes.

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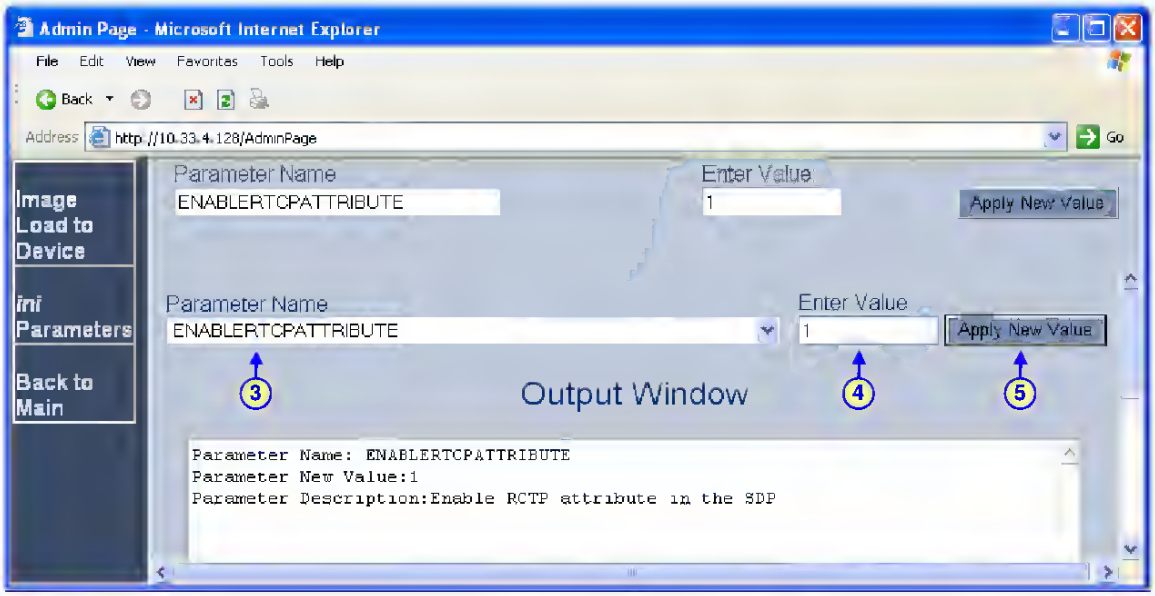
3.3.2.13 Disabling RTCP Attribute

By default, the RTP Control Protocol (RTCP) attribute (EnableRTCPAttribute) is enabled. However, you need to disable it by using the AdminPage of the Embedded Web Server.

➢ **To disable the RTCP attribute, take these 5 steps:**

1. In the Web browser's URL field, append the case-sensitive suffix 'AdminPage' to the IP address (e.g., http://10.1.229.17/AdminPage).
2. On the left pane, click **ini Parameters**, the 'ini Parameters' screen appears in the right pane.

Figure 3-23: Admin Page – ini Parameters Screen



3. From the 'Parameter Name' drop-down list, select the parameter 'ENABLERTCPATTRIBUTE'.
4. In the 'Enter Value' field, enter 1 to disable the RTCP attribute.
5. Click **Apply New Value** to apply the new setting to the gateway.

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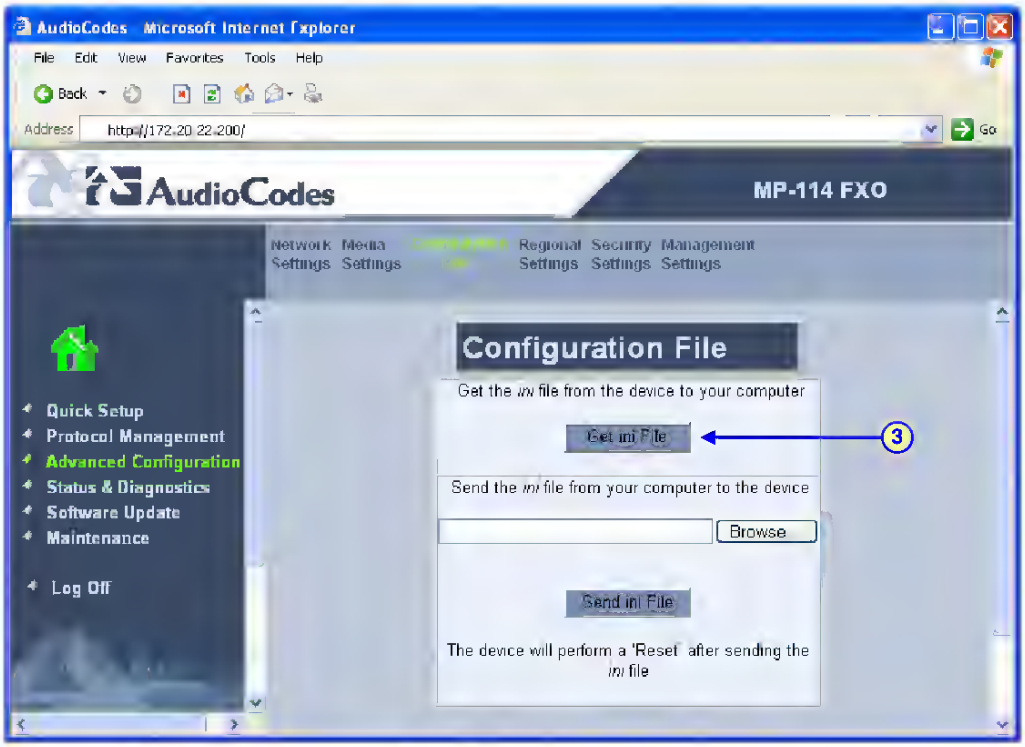
4 Backing Up Configuration Settings

Once you have modified configuration settings, it is recommended you make a backup of the modified configuration file. This is done by saving the modified *ini* configuration file to a folder on your PC. If you lose your gateway's configuration settings, you can always use this backed up *ini* file to restore them. For a description on uploading (restoring) an *ini* file to your gateway, refer to Section 3.3.1.2 on page 18.

➢ To save the *ini* file on your PC, take these 6 steps:

1. Login to the gateway's Embedded Web Server (refer to Section 3.1.1 on page 10).
2. Open the 'Configuration File' screen (**Advanced Configuration** menu > **Configuration File**).

Figure 4-1: Configuration File Screen



3. Click the **Get ini File** button, the 'File Download' window opens.
4. Click the **Save** button, the 'Save As' window opens.
5. Navigate to the folder on your PC in which you want to save the *ini* file.
6. Click the **Save** button, the gateway copies the *ini* file to the folder you selected.

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5 Monitoring the Gateway

The gateway provides several methods to monitor various operating statuses:

- Monitoring the gateway's front-panel chassis LEDs (refer to Section 5.1)
- Monitoring the gateway's channels using the Embedded Web Server (refer to Section 5.2 on page 39)

5.1 Front-Panel LEDs

The gateway provides various operating LEDs on its front-panel, as described in the table below.

Table 5-1: Description of Gateway's Front Panel LEDs

LED	Type	Color	State	Definition
Ready	Device Status	**Green**	On	Device powered, self-test OK
		Orange	Blinking	Software loading/Initialization
		Red	On	Malfunction
Uplink	Ethernet Link Status	**Green**	On	Valid 10/100 Base-TX Ethernet connection
		Red	On	No uplink
Power	Power Supply Status	**Green**	On	Power is currently being supplied to the gateway
		–	Off	Either there's a failure / disruption in the AC power supply or power is currently not being supplied to the gateway through the AC power supply entry
Fail	Failure Indication	**Red**	On	Failure (fatal error) or system initialization
		–	Off	No traffic
Channels Status	Telephone / PBX Interface	**Green**	On	The FXO gateway off-hooks the line towards the PBX
		Green	Blinking	Indicates an incoming call, before answering
		Red	On	Line malfunction
		–	Off	Normal on hook position

Note: The front panels of the MP-114 and MP-118 are similar except for the number of **Channels Status** LEDs.

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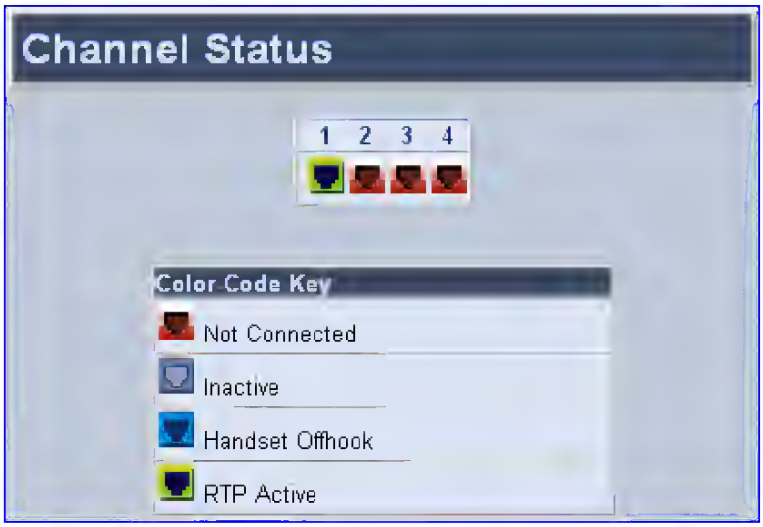
5.2 Monitoring the Channels

You can monitor channels using the gateway's Embedded Web Server

➢ To monitor the status of the channels, take this step:

- Open the 'Channel Status' screen, by clicking the Home icon on the main menu bar.

Figure 5-1: Channel Status Screen (e.g., MP-114)



The color of each channel icon indicates the call status of that channel, as described in the table below

Table 5-2: Channel Status Color Indicators

Indicator	Label	Description
	Not Connected	Indicates that no analog line is connected to this port
	Inactive	Indicates this channel is currently onhook
	Handset Offhook	Indicates this channel is offhook, but there is no active RTP session
	RTP Active	Indicates an active RTP stream

➢ To monitor the details of a channel, take these 2 steps:

1. In the 'Channel Status' screen, click the numbered channel icon of the specific channel whose detailed status you want to monitor, and then from the shortcut menu, choose **Port Settings**, the channel-specific 'Channel Status' screen appears
2. Click the submenu links to view the channel's parameter settings

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6 Troubleshooting

6.1 General Troubleshooting

The table below lists general troubleshooting that may arise with your gateway.

Table 6-1: General Troubleshooting

Problem	Possible Cause	Solution
Ready LED is lit red	Hardware component failure	Contact AudioCodes' Customer Support representative
Unable to access Embedded Web Server	Ethernet cable disconnected	Ensure that the Ethernet cable is plugged firmly into the **Ethernet** port
	No IP connectivity due to incorrect IP addressing scheme	Ensure that the IP address, subnet mask, and default gateway are correctly configured by connecting directly to the gateway using CLI (serial) or HTTP (local Ethernet connection), as described in Section 3.2 on page 14.
Configuration settings have been lost	Hardware failure.	Restore the gateway's configuration settings by uploading the backup *ini* file (refer to Section 3.3.1.2 on page 18).
Unable to access Embedded Web Server due to forgotten/lost login password	--	Restore the gateway's default configuration settings (refer to Section 6.2 on page 40). **Note:** Default username and password is 'Admin'.

6.2 Restoring Parameters to Factory Default Settings

You can use the hardware Reset button to restore the gateway's configuration parameters to their factory defaults (default IP addresses are listed in Table 3-1). Therefore, after you have restored factory defaults, you must load your previously backed-up *ini* file, or the *ini* file downloaded from AudioCodes' Web site to set the gateway's configuration parameters to their correct values.

➢ To restore factory default settings, take these 3 steps:

1. With a paper clip or any other similar pointed object, press and hold down the Reset button (located on the rear panel for MP-11x and front panel for MP-124) for about six seconds, the gateway is restored to its factory settings.
2. Assign the gateway an IP address (refer to Section 3.2 on page 14).
3. Load your previously backed-up *ini* file, or the *ini* file downloaded from the Web site. To load the *ini* file via the Embedded Web Server, refer to Section 3.3.1.2 on page 18.

6.3 Debugging using a Syslog Server

The Syslog client, embedded in the gateway, sends error reports and events generated by the gateway to a Syslog server application, using IP/UDP protocol. The Syslog server can be AudioCodes proprietary Syslog server (ACSyslog) or any third-party, Syslog server such as Kiwi Enterprises (www.kiwisyslog.com/).

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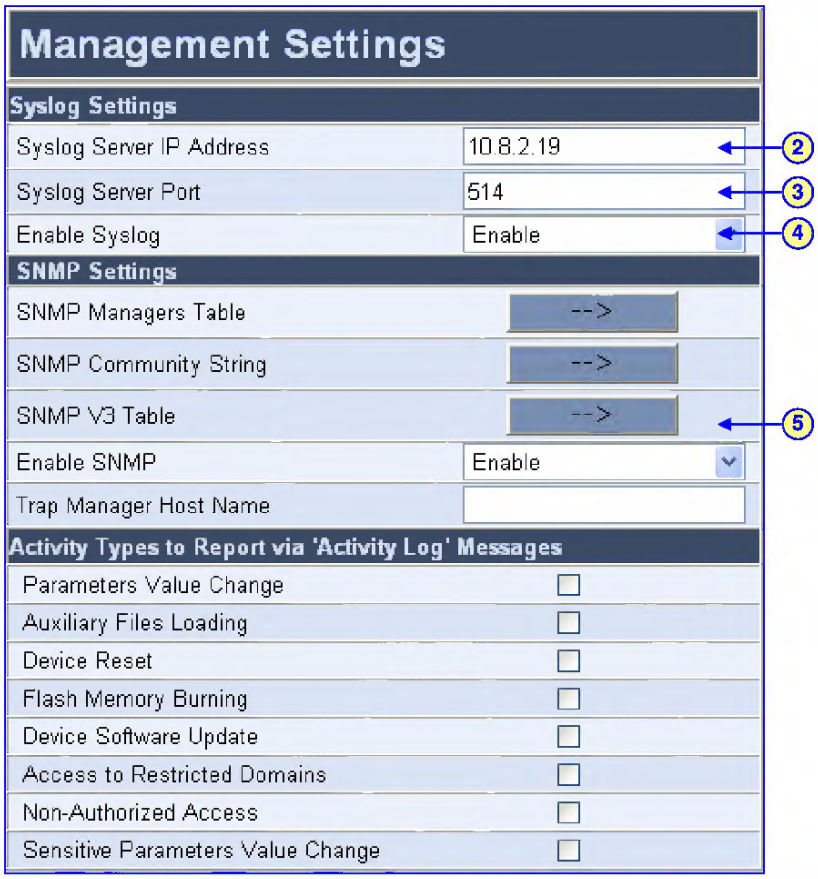
➢ To configure the Syslog parameters, take these 7 steps:

1. Open the 'Management Settings' screen (**Advanced Configuration** menu > **Management Settings**); the 'Management Settings' screen is displayed.

Figure 6-1: Management Settings Screen



2. In the 'Syslog Server IP Address' field, enter the Syslog's IP address. Note that the Syslog server's IP address must correspond to the network environment in which the Syslog server is installed.
3. In the 'Syslog Server Port' field, enter the UDP port number (default port is 514).
4. From the 'Enable Syslog' drop-down list, enable the Syslog feature by selecting 'Enable'.
5. In the 'Activity Types to Report' group, select the Web operations that you want logged to the Syslog server.
6. To define the messages based on severity level that you want sent to the Syslog server, open the 'General Parameters' screen (**Protocol Management** menu > **Advanced Parameters** > **General Parameters**), and then from the 'Debug Level' drop-down list, select '5' (i.e., all reports with severity level 5 are sent to the Syslog server).
7. To enable additional call information sent to the Syslog, in the 'CDR Report Level' drop-down list, select 'End Call'

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7 Regulatory Information

Declaration of Conformity

Application of Council Directives:	73/23/EEC (including amendments)	
	89/336/EEC (including amendments)	
	1999/5/EC Annex-II of the Directive	
Standards to which Conformity is Declared:	EN55022: 1998 + A1: 2000 + A2: 2003	
	EN55024:1998 + A1: 2001 + A2: 2003	
	EN61000-3-2: 2000 + A2: 2005	
	EN61000-3-3: 1995 + A1: 2001	
	EN60950-1: 2001	
Manufacturer's Name:	AudioCodes Ltd.	
Manufacturer's Address:	1 Hayarden Street, Airport City, Lod 70151, Israel.	
Type of Equipment:	Analog VoIP System	
Model Numbers:	**MP-11x/FXS+FXO Mixed Series:**	MP-114/ 2FXS/2FXO; MP-118/ 4FXS/4FXO
	MP-11x/FXO Series:	MP-112/ 2FXO; MP-114/ 4FXO; MP-118/ 8FXO

I, the undersigned, hereby declare that the equipment specified above conforms to the above Directives and Standards.

	27th June, 2006	Airport City, Lod, Israel
Signature	*Date (Day/Month/Year)*	*Location*

I. Zusmanovich, Compliance Engineering Manager

Czech	[AudioCodes Ltd] tímto prohlašuje, že tento [MP-11x/FXO] je ve shodě se základními požadavky a dalšími příslušnými ustanoveními směrnice 1999/5/ES."
Danish	Undertegnede [AudioCodes Ltd] erklærer herved, at følgende udstyr [MP-11x/FXO] overholder de væsentlige krav og øvrige relevante krav i direktiv 1999/5/EF
Dutch	Hierbij verklaart [AudioCodes Ltd] dat het toestel [MP-11x/FXO] in overeenstemming is met de essentiële eisen en de andere relevante bepalingen van richtlijn 1999/5/EG
English	Hereby, [AudioCodes Ltd], declares that this this [MP-11x/FXO] is in compliance with the essential requirements and other relevant provisions of Directive 1999/5/EC.
Estonian	Käesolevaga kinnitab [AudioCodes Ltd] seadme [MP-11x/FXO] vastavust direktiivi 1999/5/EÜ põhinõuetele ja nimetatud direktiivist tulenevatele teistele asjakohastele sätetele.
Finnish	[AudioCodes Ltd] vakuuttaa täten että [MP-11x/FXO] tyyppinen laite on direktiivin 1999/5/EY oleellisten vaatimusten ja sitä koskevien direktiivin muiden ehtojen mukainen.
French	Par la présente [AudioCodes Ltd] déclare que l'appareil [MP-11x/FXO] est conforme aux exigences essentielles et aux autres dispositions pertinentes de la directive 1999/5/CE
German	Hiermit erklärt [AudioCodes Ltd], dass sich dieser/diese/dieses [MP-11x/FXO] in Übereinstimmung mit den grundlegenden Anforderungen und den anderen relevanten Vorschriften der Richtlinie 1999/5/EG befindet". (BMWi)
Greek	ΜΕ ΤΗΝ ΠΑΡΟΥΣΑ [AudioCodes Ltd] ΔΗΛΩΝΕΙ ΟΤΙ [MP-11x/FXO] ΣΥΜΜΟΡΦΩΝΕΤΑΙ ΠΡΟΣ ΤΙΣ ΟΥΣΙΩΔΕΙΣ ΑΠΑΙΤΗΣΕΙΣ ΚΑΙ ΤΙΣ ΛΟΙΠΕΣ ΣΧΕΤΙΚΕΣ ΔΙΑΤΑΞΕΙΣ ΤΗΣ ΟΔΗΓΙΑΣ 1999/5/ΕΚ
Hungarian	Alulírott, [AudioCodes Ltd] nyilatkozom, hogy a [MP-11x/FXO] megfelel a vonatkozó alapvető követelményeknek és az 1999/5/EC irányelv egyéb előírásainak
Icelandic	æki þetta er í samræmi við tilskipun Evrópusambandsins 1999/5
Italian	Con la presente [AudioCodes Ltd] dichiara che questo [MP-11x/FXO] è conforme ai requisiti essenziali ed alle altre disposizioni pertinenti stabilite dalla direttiva 1999/5/CE.
Latvian	Ar šo [AudioCodes Ltd] deklarē, ka [MP-11x/FXO] atbilst Direktīvas 1999/5/EK būtiskajām prasībām un citiem ar to saistītajiem noteikumiem.
Lithuanian	[AudioCodes Ltd] deklaruoja, kad įrenginys [MP-11x/FXO] tenkina 1999/5/EB Direktyvos esminius reikalavimus ir kitas šios direktyvos nuostatas
Maltese	Hawnhekk, [AudioCodes Ltd], jiddikjara li dan [MP-11x/FXO] jikkonforma mal-ħtiġijiet essenzjali u ma provvedimenti oħrajn relevanti li hemm fid-Direttiva 1999/5/EC
Norwegian	Dette produktet er i samhørighet med det Europeiske Direktiv 1999/5
Polish	[AudioCodes Ltd], deklarujemy z pełną odpowiedzialnością, że wyrób [MP-11x/FXO] spełnia podstawowe wymagania i odpowiada warunkom zawartym w dyrektywie 1999/5/EC
Portuguese	[AudioCodes Ltd] declara que este [MP-11x/FXO] está conforme com os requisitos essenciais e outras disposições da Directiva 1999/5/CE.
Slovak	[AudioCodes Ltd] týmto vyhlasuje, že [MP-11x/FXO] spĺňa základné požiadavky a všetky príslušné ustanovenia Smernice 1999/5/ES.
Slovene	Šiuo [AudioCodes Ltd] deklaruoja, kad šis [MP-11x/FXO] atitinka esminius reikalavimus ir kitas 1999/5/EB Direktyvos nuostatas.
Spanish	Por medio de la presente [AudioCodes Ltd] declara que el [MP-11x/FXO] cumple con los requisitos esenciales y cualesquiera otras disposiciones aplicables o exigibles de la Directiva 1999/5/CE
Swedish	Härmed intygar [AudioCodes Ltd] att denna [MP-11x/FXO] står I överensstämmelse med de väsentliga egenskapskrav och övriga relevanta bestämmelser som framgår av direktiv 1999/5/EG.

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Safety Notices

- Installation and service of this unit must only be performed by authorized, qualified service personnel.
- The MediaPack FXO Output Tones and DTMF level should not exceed -9 dBm (AudioCodes setting #23) in order to comply with FCC 68, TIA/EIA/IS-968 and TBR-21.
- The maximum allowed gain between any 2 ports connected to the PSTN should be set to 0 dB in order to comply with FCC 68, TIA/EIA/IS-968 Signal power limitation.

Industry Canada Notice

This equipment meets the applicable Industry Canada Terminal Equipment technical specifications. This is confirmed by the registration numbers. The abbreviation, IC, before the registration number signifies that registration was performed based on a declaration of conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

The Ringer Equivalence Number (REN) for this terminal is 0.5. The REN assigned to each terminal equipment provides an indication of the maximum number of terminals allowed to be connected to a telephone interface. The termination on an interface may consist of any combination of devices subject only to the requirement that the sum of Ringer Equivalence Number of all devices do not exceed five.

Network Compatibility

The products support the Telecom networks in EU that comply with TBR21.

Telecommunication Safety

The safety status of each port is declared and detailed in the table below:

Ports	Safety Status
Ethernet (100 Base-TX)	SELV
FXO	TNV-3

TNV-3: Circuit whose normal operating voltages exceeds the limits for an SELV circuit under normal operating conditions and on which over voltages from Telecommunication Networks are possible.

SELV: Safety extra low voltage circuit.

FCC Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

ACTA Customer information

1. This equipment, the VoIP Analog Gateway, models MP-118, MP-114 and MP-112 complies with Part 68 of the FCC Rules and the requirements adopted by the ACTA. On the bottom of the unit of this equipment is a label, that contains among other information, a product identifier in the format US:AC1IT00BMP11X3AC. If requested, this number must be provided to the telephone company.
2. This equipment is designed to be connected to the telephone network using an RJ-11C connector, which is Part 68 compliant. The service order code (SOC) is 9.0Y and the Facility interface code (FIC) is 02LS2.
3. The REN is used to determine the number of devices that may be connected to a telephone line. Excessive RENs on a telephone line may result in the devices not ringing in response to an incoming call. In most but not all areas, the sum of RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company. The REN for this product is part of the product identifier that has the format US:AC1IT00BMP11X3AC The digits represented by 00 are the REN without a decimal point.
4. Should the product causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. If advance notice is not practical, you will be notified as soon as possible. Also, you will be advised of your right to file a compliant with the FCC if it is necessary.
5. The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.
6. If trouble is experienced with this equipment, for repair or warranty information please contact AudioCodes Inc., 2099 Gateway Place, Suite 500, San Jose, CA, 95110, phone number 1-408-441-1175. If the equipment is causing harm to the telephone network, the telephone company may request to disconnect the equipment until the problem is resolved.
7. Installation is described in the Product User's manual. Connection to Telephone Company-provided coin service is prohibited. Connection to party lines service is subject to State tariffs.

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AudioCodes™

MediaPack MP-118 & MP-114

Quick Installation Guide

Integration with Microsoft® Office Communications Server 2007

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